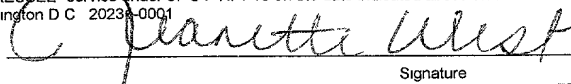| FORM PTO-1390 DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE (REV 5-93)	ATTORNEY'S DOCKET NO. 851663.407
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371	U.S. APPLICATION NO. (If known, see 37 CFR 1.5) Unknown **09/486582**

INTERNATIONAL APPLICATION NO. PCT/SG97/00037	INTERNATIONAL FILING DATE 29 August 1997 (29.08.1997)	PRIORITY DATE CLAIMED 29 August 1997 (29.08.1997)

TITLE OF INVENTION
FAST SYNTHESIS SUB-BAND FILTERING METHOD FOR DIGITAL SIGNAL DECODING

APPLICANT(S) FOR DO/EO/US
SAPNA, George; and YANG, Haiyun

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.

2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.

3. ☒ This express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).

4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.

5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2)).

    a. ☒ is transmitted herewith (required only if not transmitted by the International bureau).

    b. ☐ has been transmitted by the International Bureau.

    c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).

6. ☐ A translation of the International Application into English (35 U.S.C. 371(c)(2)).

7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3)).

    a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).

    b. ☐ have been transmitted by the International Bureau.

    c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.

    d. ☒ have not been made and will not be made.

8. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).

9. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)).

10. ☐ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

Items 11 to 16 below concern document(s) or information included:

11. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.

12. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.

13. ☐ A FIRST preliminary amendment.
    ☐ A SECOND or SUBSEQUENT preliminary amendment.

14. ☐ A substitute specification.

15. ☐ A change of power of attorney and/or address letter.

16. ☒ Other items or information:

    Article 34 Amendments, Postcard, and check for filing fees.

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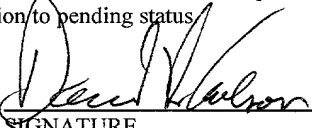
NUMBER **EL487802128US**

DATE OF DEPOSIT **28 February 2000**

I hereby certify that this paper or fee is being deposited with the United States Postal Service "EXPRESS MAIL POST OFFICE TO ADDRESSEE" service under 37 C F R. 1 10 on the date indicated above and is addressed to· Assistant Commissioner for Patents, Washington D C 20231-0001

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U.S. APPLICATION NO. (If known, see 37 CFR 1.5) Unknown 09/486582	INTERNATIONAL APPLICATION NO. PCT/SG97/00037	ATTORNEY'S DOCKET NUMBER 851663.407

17. ☒ The following fees are submitted:

Basic National Fee (37 CFR 1.492(a)(1)-(5)):

	CALCULATIONS PTO USE ONLY

Search Report has been prepared by the EPO or JPO ....$ 840.00

International preliminary examination fee paid to USPTO (37 CFR 1.482)
....$ 670.00

No international preliminary examination fee paid to USPTO (cu CFR 1.482)
but international search fee paid to USPTO (37 CFR 1.445(a)(2))...................$ 690.00

Neither international preliminary examination fee (37 CFR 1.482) nor
international search fee (37 CFR 1.445(a)(2)) paid to USPTO $970.00

International preliminary examination fee paid to USPTO (37 CFR 1.482)
and all claims satisfied provisions of PCT Article 33(2)-(4).............................$ 96.00

		CALCULATIONS (PTO USE ONLY)	
ENTER APPROPRIATE BASIC FEE AMOUNT	**=**	$840.00	
Surcharge of $130.00 for furnishing the oath or declaration later than ☐ 20 ☒ 30 months from the earliest claimed priority date (37 CFR 1.492(e)).		$.00	

Claims	Number Filed	Number Extra	Rate		
Total Claims	17 - 20 =	0	x $ 18.00	$0.00	
Independent Claims	3 - 3 =	0	x $ 78.00	$0.00	
Multiple dependent claim(s) (if applicable)			+ $260.00	$260.00	

TOTAL OF ABOVE CALCULATIONS	**=**	$1100.00	
Reduction by ½ for filing by small entity, if applicable. Verified Small Entity statement must also be filed. (NOTE: 37 CFR 1.9, 1.27, 1.28)		$.00	
SUBTOTAL	**=**	$1100.00	
Processing fee of $130.00 for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(f)). +		$.00	
TOTAL NATIONAL FEE	**=**	$1100.00	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31). $40.00 per property) +		$.00	
TOTAL FEES ENCLOSED	**=**	**$1100.00**	
		Amount to be refunded:	
		charged	

a. ☒ A check in the amount of $1100.00 cover the above fees is enclosed.

b. ☐ Please charge my Deposit Account No. in the amount of $_ to cover the above fees. A duplicate copy of this sheet is enclosed.

c. ☒ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. **19-1090**. A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

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31,153
REGISTRATION NUMBER

- 1 -

FAST SYNTHESIS SUB-BAND FILTERING FOR DIGITAL SIGNAL DECODING

This invention relates to digital signal decoding for the purposes primarily of audio reproduction. In particular, the invention relates to enhanced synthesis sub-band filtering

5 during decoding of digital audio signals.

In order to store or transmit data representing audio signals it is often desirable to first encode or compress the data so as to enable it to be stored or transmitted more efficiently. Decoding the data requires that the stored or transmitted data be reconstructed into audio signals by

10 application of a decoding or decompression technique. The reconstruction process is typically quite computationally intensive, yet the process should be fast and reliable enough to enable the audio signals to be reconstructed in real time, on the fly, for example. In order for the decoding process to be carried out in relatively low-cost consumer products, the hardware utilised by the decoder should also preferably be relatively simple and inexpensive, or at least

15 to the greatest extent reasonably possible.

Efficient stereo and multichannel digital audio signal coding methods have been developed for storage or transmission applications such as Digital Audio Broadcasting (DAB), Integrated Service Digital Network (ISDN), High Definition Television (HDTV) and Set Top Box (STB)

20 for video-on-demand. The formats used to encode and reciprocally decode digital audio and video information for storage and retrieval is subject to various standards, one of which has been established by the Moving Pictures Experts Group and is known as the MPEG standard. A standard on low bit rate coding for mono or stereo audio signals was established by MPEG-1 Audio, published under ISO-IEC/JTC1 SC29 11172-3, entitled "Coding of Moving Pictures

25 and Associated Audio for Digital Storage Media at up to About 1.5 Mbit/s", and the disclosure of that document is incorporated herein by reference. MPEG-2 Audio (ISO/IEC 13818-3) provides the extension to 3/2 multichannel audio and an optional low frequency enhancement channel (LFE). The audio part of the standard, ISO/IEC 11172-3, defines three algorithms, Layer 1, 2 and 3 for coding PCM audio signals. MPEG-2 (Multichannel) also

30 defines Layer 1, 2 and 3 algorithms.

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The MPEG audio encoder processes a digital audio signal and produces a compressed bitstream for transmission or storage. The encoder algorithm is not standardised, and may use various means for encoding such as estimation of the auditory masking threshold, quantisation, and scaling. However, the encoder output must be such that a decoder 5 conforming to the above-mentioned standards specification will produce audio suitable for the intended application.

The decoder, subject to the application-dependent parameters, accepts the compressed audio bitstream in the defined syntax, decodes the data elements and uses the information to produce 10 digital audio output, also according to the defined standard. The decoder first unpacks the received bitstream to recover the encoded audio information frame by frame. After the process of frame unpacking, the decoder performs an inverse quantisation (expansion process) and feeds a sub-band synthesis filter bank with a set of 32 scaled-up sub-band samples in order to reconstruct the output PCM audio signals. The sub-band filter banks used for Layer 15 1 and Layer 2 of MPEG 1 audio decoder and Layer 1 and Layer 2 of MPEG2 (Multichannel extension) audio decoder, are the same.

The sub-band synthesis filter is one of the most computationally intensive blocks of the MPEG audio decoder. Sub-band filtering is performed for each sub-band in a frame and for every 20 channel. Any reduction in its computational requirements thus enables less complexity and reduced cost of decoding.

In accordance with the present invention there is provided a method of decoding digital audio data, comprising the steps of obtaining an input sequence of data elements representing 25 encoded audio samples, calculating an array of sum data and an array of difference data using selected data elements from the input sequence, calculating a first sequence of output values using the array of sum data, calculating a second sequence of output values using the array of difference data, and forming decoded audio signals from the first and second sequences of output data.

30

Preferably, the array of sum data is obtained by adding together respective first and second

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data elements from the input sequence, the first and second data elements being selected from mutually exclusive sub-sequences of the input sequence. Furthermore, the array of difference data is preferably obtained by subtracting respective first data elements from corresponding second data elements of the input sequence, the first and second data elements being selected

5 from mutually exclusive sub-sequences of the input sequence.

In one form of the invention the step of calculating an array of sum data and an array of difference data comprises dividing the input data sequence into first and second equal sized sub-sequences, the first sub-sequence comprising the high order data elements of the input

10 sequence and the second sub-sequence comprising the low order data elements of the input sequence, calculating the array of sum data by adding together each respective data element of the first sub-sequence with a respective corresponding data element of the second sub-sequence, and calculating the array of difference data by subtracting each respective data element of the first sub-sequence from a respective corresponding data element of the second

15 sub-sequence.

The invention also provides method of decoding a sequence of m, m an even positive integer, input digital audio data samples $S[k]$, where $k = 0, 1, ... (m-1)$, to produce a set of n, n an even positive integer, output audio data samples $V[i]$, where $i = 0, 1, ...(n-1)$, comprising

20 the steps of:

a) calculating an array of sum data $S_{ADD}[k]$ according to

$$S_{ADD}[k] = S[k] + S[m-1-k] \qquad \text{for } k = 0, 1, ...(m/2-1)$$

b) calculating an array of difference data $S_{SUB}[k]$ according to

$$S_{SUB}[k] = S[k] - S[m-1-k] \qquad \text{for } k = 0, 1, ...(m/2-1)$$

25 c) calculating a first output audio data sample by a multiply-accumulate operation according to

$$V[2i] = V[2i] + N[2i, k]*S_{ADD}[k] \qquad \text{for } k = 0, 1, ... (m/2-1)$$
$$\text{where } N[2i, k] = \cos\left[\frac{(32+2i)(2k+1)\pi}{64}\right]$$

d) calculating a second output audio data sample by a multiply-accumulate operation

30 according to

$$V[2i+1] = V[2i+1] + N[2i+1, k]*S_{SUB}[k] \qquad \text{for } k = 0, 1, ... (m/2-1)$$

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where $N[2i+1, k] = \cos\left[\dfrac{(32+(2i+1))(2k+1)\pi}{64}\right]$

e)   and repeating steps c) and d) for i = 0, 1, ... (n/2-1) to obtain a full set of output data.

The invention further provides a synthesis sub-band filter for use in decoding digital audio
5   data, comprising a means for receiving or retrieving an input sequence of data elements
comprising encoded digital audio data, a pre-calculation means for calculating an array of sum
data and an array of difference data using selected data elements from the input sequence, and
a transform calculation means for calculating a first sequence of decoded output values using
said array of sum data and a second sequence of decoded output values using said array of
10   difference data.

The invention is described in greater detail hereinbelow, by way of example only, with
reference to the accompanying drawings, in which:

Figure 1 is a block diagram of major functional portions of an MPEG audio encoder;
15   Figure 2 is a block diagram of major functional portions of an MPEG audio decoder;

Figure 3 is a flow diagram of an MPEG decoding procedure;

Figure 4 is a flow diagram showing a generalised form of a procedure according to
the present invention; and

Figure 5 is a flow diagram illustrating a preferred implementation of the invention.
20

Figure 1 is a block diagram illustrating the major components of an MPEG audio encoder
circuit 2 constructed in accordance with the aforementioned standards document. In the
figure, an input signal 4, comprising a pulse code modulated (PCM) signal having a 48 kHz
sampling frequency and a sample size of 16 bits per sample, is provided as input to the single
25   channel encoder 2. The input signal is first mapped from the time domain into the frequency
domain by a sub-band filter bank 8. The resulting coefficients are normalized with scale
factors which may be transmitted as side information. The coefficients thus obtained are then
quantized and entropy encoded by a quantizer and encoding circuit 10. Masking thresholds
of the quantization errors are calculated based on psychoacoustic values provided by a
30   psychoacoustic model 14 to control the quantization step. The bit allocation is transmitted as
side information. The coded signal is then multiplexed by a frame packing circuit 12 and an

- 5 -

encoded bitstream 6 is produced at the output of the encoder 2.

A block diagram illustrating the main components of an MPEG audio decoder circuit 20 is shown in Figure 2. In the figure, an encoded bitstream 22 is provided to the input of the
5 decoder. A bitstream unpacking and decoding circuit 26 performs an error correction operation if such operation was applied in the encoder. The bitstream data are unpacked to recover the various pieces of encoded information, and a reconstruction circuit 28 reconstructs the quantized version of the set of mapped samples from the frames of input data. An inverse mapping circuit 30 transforms the mapped samples back into a uniform pulse code modulated
10 (PCM) output signal 24 that reproduces the corresponding input signal which was provided to the encoder.

The foregoing descriptions of the encoder and decoder are specific to the MPEG standard, and it is considered to be within the skill of those in the art to implement the various hardware
15 functions described above. Accordingly, a more detailed hardware description of an MPEG coding system is not considered necessary for a full and complete understanding of the invention. It should be appreciated the invention described herein, although described in connection with the MPEG coding standard, is considered useful for other coding applications and standards.
20
Referring to Figure 3, there is shown a flow diagram 40 of steps involved in signal processing in layers I and II in an MPEG1 audio decoder. To begin with, the bit allocation of an input bitstream (42, 44) is decoded (46). Thereafter, various scale factors are also decoded (48) and the samples are requantized (50). The encoded signal is decoded in a synthesis sub-band filter
25 (52) and the decoded pulse code modulated signals are output (54, 56) for further processing and/or real time reproduction. The present invention relates primarily to the synthesis sub-band filter portion of the decoding process, when implemented for MPEG decoding.

The synthesis sub-band filter bank is composed of two main functions, an Inverse Modified
30 Discrete Cosine Transform (IMDCT) and an Inverse Pseudo-Quadrature Mirror Filter (IPQMF). The IMDCT, which can be viewed as an overlap transform, performs a 32 x 64

cosine modulation transformation, which means a frequency shift of a filter bank into one single filter.

Consider a system in which output sub-band audio signal samples V_i (i=0....63) are decoded
5 from sequences of 32 encoded input samples S_k, k = 0....31. The inverse MDCT of the sequence S_k, is defined as follows:

$$V_i = \sum_{k=0}^{31} \cos\left[\frac{(16+i)(2k+1)\pi}{64}\right] * S_k \qquad (1)$$
$$for \ i=0,1,.....63$$

Taking the cosine symmetric property wherein:

$$\cos\theta = \cos(2\pi - \theta) \qquad (2)$$

10 the IMDCT definition equation (1) may be modified as given below:

$$V_i = \sum_{k=0}^{15} \cos\left[\frac{(32+i)(2k+1)\pi}{64}\right] * \left[S_k + (-1)^i * S_{31-k}\right] \qquad (3)$$
$$for \ i=0,1,.....31$$

This equation (3) may be computed according to the following algorithm:

repeat i = 32 times
15   *repeat k = 16 times*
    if I is even, Sum = S[k] + S[31-k]
    if I is odd, Sum = S[k] - S[31-k]
    *V[i] = V[i] + N[i, k] * Sum*
  end k
20   *end i*

where i *is the index of output samples (i=0....31)*

            k *is the index of input samples (k=0....15)*

$$N(i,k) = \cos\left[\frac{(32+i)(2k+1)\pi}{64}\right]$$

            $S[k]$ *represents the input sample data sequence*

5           $V[i]$ *represents the output of IMDCT*

The IMDCT equation, making use of the symmetrical property, is given in Equation (3) above, and the computational effort required for MPEG audio decoding is in large part dependant upon the efficiency with which the input samples can be processed through the

10 IMDCT to obtain respective sub-band filter PCM samples. Embodiments of the present invention are able to reduce the number of arithmetic operations performed in implementing the IMDCT portion of the decoder, to thereby increase the computational efficiency of the decoding process. In particular, the number of addition operations required for the implementation of this equation can be reduced substantially by pre-computing the sum and

15 difference of the sample data which is the input to the IMDCT. In addition, the pre-computation can take place outside the main IMDCT computational loop. Hence the main loop contains only the MAC operations, which can be executed very efficiently by any general purpose DSP in a minimum number of cycles.

20 In the present invention, the dequantised sample data (e.g. 32 samples) from the encoded bitstream is pre-processed as per the symmetrical property of the cosine coefficients. The sample data is then split into two banks, each containing 16 samples. The sum and difference of respective data elements in the two banks is computed and stored in two arrays. These arrays are used as the input data for the subsequent MAC operations.

25

Prior art implementations of equation (3) have required 32 x 16 Multiply-Accumulate operations and 32 x 16 Addition operations. By using the pre-computation operations described above, however, the number of Addition operations reduces to 2 x 16. This results in a saving of 30 x 16 Addition operations per Sub-band filter implementation, which in turn

30 translates to a corresponding reduction in overall computational power.

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In the IMDCT equation (3), S_k represents a sequence of m input data samples, where k = 0 ... (m-1). In a typical implementation for MPEG decoding 32 input data samples may be processed, such that m=32. For pre-computing the sum and difference of respective data elements, the input data sample sequence is first arranged into two equally sized data banks,

5   one constituting the high order data elements and the other the low order data elements:

Data bank (1) S_k for k = 0 ... (m/2)-1

Data bank (2)   Sk   for k = (m/2) ... (m-1)

For example, in a preferred embodiment of the present invention where m=32, S_k is split into

10 two data banks comprising:

(1) S_k for k = 0 .. 15

(2) S_k for k = 16 .. 31

The sum and difference data are calculated using respective data elements from the two data

15 banks and is stored in two arrays of data, S_{ADD} and S_{SUB}, which are computed as follows:

$$S_{ADD}[k] = S[k] + S[m-1-k] \qquad for \ k = 0, 1, ...(m/2)-1 \qquad (4)$$

$$S_{SUB}[k] = S[k] - S[m-1-k] \qquad for \ k = 0, 1, ...(m/2)-1 \qquad (5)$$

In the aforementioned example of 32 input data samples, equations (4) and (5) reduce to:

$S_{ADD}[k] = S[k] + S[31-k]$ \qquad for k = 0, 1, ...15

$S_{SUB}[k] = S[k] - S[31-k]$ \qquad for k = 0, 1, ...15

20

The IMDCT equation (3) may now be divided into two portions and rewritten as follows:

$$V[i] = \sum_{k=0}^{15} \cos\frac{(32+i)(2k+1)\pi}{64} * S_{ADD}[k]$$
$$for \ i = 0,2,4,....30 \qquad (6)$$

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$$V[i] = \sum_{k=0}^{15} \cos\frac{(32+i)(2k+1)\pi}{64} * S_{SUB}[k] \qquad (7)$$

$$for \ i = 1,3,5,.....31$$

As shown in the above equations (6) and (7), the IMDCT may now be calculated in two passes, an 'even pass' where the sum of the sample data is used (equation (6)), and an 'odd pass' where the difference of the sample data is used (equation (7)). The computational algorithms of the above equations are shown below.

5

Calculation of sum and difference of sample data (Addition operations)

  repeat k = 16 times

  $S_{ADD}[k] = S_k + S_{31-k}$

  $S_{SUB}[k] = S_k - S_{31-k}$

10 *end k*

Calculation of 'even' data of IMDCT (Multiply-Accumulate operations)

  repeat i = 16 times

  repeat k = 16 times

15 $V[i] = V[i] + N[i,k]*S_{ADD}[k]$

  end k

  end i

Calculation of 'odd' data of IMDCT (Multiply-Accumulate operations)

20 *repeat i = 16 times*

  repeat k = 16 times

  $V[i] = V[i] + N[i,k]*S_{SUB}[k]$

  end k

  end   i

25

where   *i*   *is the index of output samples (i=0...31)*

          k   *is the index of input samples (k=0....15)*

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$$N(i,k) = \cos\left[\frac{(32+i)(2k+1)\pi}{64}\right]$$

$S[k]$ *represents the input sample data sequence*

S_{ADD} *represents the sum of data array*

S_{SUB} *represents the difference of data array*

5          $V[i]$ *represents the output of the IMDCT*

Figures 4 and 5 illustrate the above procedure according to a preferred embodiment of the invention in the form of flow diagrams. The representation shown in Figure 4, illustrates the general steps involved, and the procedure illustrated in the flow diagram 80 of Figure 4

10 corresponds to the synthesis sub-band filter step 52 of the overall decoding procedure 40 of Figure 3. To begin with the input samples S_k are received (82, 84) after having been isolated from the frames of encoded data received or retrieved. The input data samples are then utilised for pre-calculation of sum and difference data (step 86), as described above. This involves dividing the input data sample set into two equal sized sub-sets, which in the

15 preferred embodiment consists of a first sub-set comprising the lower order data and a second sub-set comprising the higher order data. For example, in the case of 32 input samples S_0 to S_{32} as described, the first sub-set of input sample data may comprise the lower order input data S_0 to S_{15} and the second sub-set comprises the upper order data samples S_{16} to S_{31}. Respective ones of each sub-set of input sample data are then used to obtain a sets of sum and

20 difference data, S_{ADD} and S_{SUB}. As can be readily ascertained from the above description, in the preferred embodiment the calculation of the sum and difference data is performed using the lowest order samples from the first set with the corresponding highest samples from the second set. For example, in the case of 32 input samples, the sum and difference data elements may be calculated as follows:

25

$S_{ADD}[0] = S[0] + S[31]$ $S_{SUB}[0] = S[0] - S[31]$

$S_{ADD}[1] = S[1] + S[30]$ $S_{SUB}[1] = S[1] - S[30]$

$S_{ADD}[2] = S[2] + S[29]$ $S_{SUB}[2] = S[2] - S[29]$

               :   :

30   :   :

$S_{ADD}[15] = S[15] + S[16]$ $S_{SUB}[15] = S[15] - S[16]$

Once the arrays of sum and difference data have been calculated, the multiply-accumulate operations required to calculate the IMDCT can be performed iteratively in two steps. The first step (88) is used to obtain half of the output samples (e.g. the "even" outputs) using the pre-calculated sum data comprising the S_{ADD} data elements. The second step (90) is used to

5 obtain the other half of the output samples (e.g. the "odd" outputs) using the pre-calculated difference data comprising the S_{SUB} data elements. Each of these steps (88, 90) is an iterative multiply-accumulate (MAC) operation involving each of the data elements from the respective S_{ADD} or S_{SUB} array. Furthermore, each of the MAC operations of steps 88, 90 are performed repeatedly (step 92) to obtain a full complement of output samples. For example, where 32

10 output samples V_0 to V_{32} are required, each of the iterative MAC steps 88, 90 would be performed 16 times. Once the data for each output has been calculated, the data samples are output for PCM processing (step 94).

A more detailed preferred embodiment of the decoding procedure is illustrated in the flow

15 diagram 100 shown in Figure 5. Beginning at step 102, a sequence of m input samples S_k (k = 0 .... m-1) are received for decoding to n sub-band filter outputs V_i (i = 0 .... n-1) at step 104. In the preferred embodiment for an MPEG implementation, both the number of input samples m and the number of output samples n are the same, 32. Steps 106, 108 and 110 of procedure 100 for a loop for the pre-calculation process of determining and storing the sum

20 and difference data arrays from the input data samples. The steps 112, 114, and 116 then form nested loops for the iterative multiple-accumulate calculation of the "even" ones of the output data elements (e.g. V_i for i = 0, 2, 4, ... 14), using the pre-calculated sum data array S_{ADD}. A calculation loop of steps 112 and 114 provides the iterative MAC operation, whilst the loop provided by step 116, enables calculation of each (even) alternate output data

25 element. The remaining (odd) alternate output data elements are calculated in nested loop steps 118, 120, 122 using the difference data array S_{SUB}. The resulting output sub-band data is then provided at final step 124.

The preferred form of the invention presented herein results in a reduction of 480 addition

30 operations per 32 sub-band samples. For a stereo output MPEG audio decoder, this is a reduction of 480 *36*2 arithmetic operations per frame. The overall reduction in arithmetic

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operations which is achieved is approximately 46.875% per IMDCT.

It will be readily apparent to those of ordinary skill in the relevant art that the present invention may be implemented in numerous different ways, without departing from the spirit
5 and scope of the invention as described herein, and it is to be understood that such modifications are considered to be within the scope of the invention. In any event, it is immediately recognisable that one way the invention can be carried out, relating as it does to the processing of data, is using general purpose computing apparatus operating under the instruction of software or the like which is produced separately and specially adapted to
10 perform the methods of the invention. Alternatively, specialised computing apparatus such as a dedicated integrated circuit, chipset or the like may be constructed with the functions of the invention embedded therein. Many other variations to the particular implementation will of course be possible. It will also be recognised that in places in the description and appended claims where it is said that a data set is divided into sub-sets, for example, this division may
15 be simply a notional one, and no physical separation need occur, as is known in the data processing art.

The foregoing detailed description of the present invention has been presented by way of example only, and is not intended to be considered limiting to the invention which is defined
20 in the claims appended hereto.

CLAIMS:

1.   A method of decoding digital audio data, comprising the steps of obtaining an input
sequence of data elements representing encoded audio samples, preprocessing the input
5 sequence of data elements by calculating an array of sum data and an array of difference data
using selected data elements from the input sequence, calculating a first sequence of output
values using the array of sum data, calculating a second sequence of output values using the
array of difference data, and forming decoded audio signals from the first and second
sequences of output values.

10

2.   A method as claimed in claim 1, wherein the array of sum data is obtained by adding
together respective first and second data elements from the input sequence, the first and
second data elements being selected from mutually exclusive sub-sequences of the input
sequence.

15

3.   A method as claimed in claim 1 or 2, wherein the array of difference data is obtained
by subtracting respective first data elements from corresponding second data elements of the
input sequence, the first and second data elements being selected from mutually exclusive sub-
sequences of the input sequence.

20

4.   A method as claimed in claim 1, wherein the step of calculating an array of sum data
and an array of difference data comprises dividing the input data sequence into first and
second equal sized sub-sequences, the first sub-sequence comprising the high order data
elements of the input sequence and the second sub-sequence comprising the low order data
25 elements of the input sequence, calculating the array of sum data by adding together each
respective data element of the first sub-sequence with a respective corresponding data element
of the second sub-sequence, and calculating the array of difference data by subtracting each
respective data element of the first sub-sequence from a respective corresponding data element
of the second sub-sequence.

30

5.   A method as claimed in claim 1, wherein the step of calculating a first sequence of

output values comprises performing a multiply-accumulate operation utilising each of the sum data elements.

6.   A method as claimed in claim 1 or 5, wherein the step of calculating a second

5   sequence of output values comprises performing a multiply-accumulate operation utilising each of the difference data elements.

7.   A method as claimed in any preceding claim wherein the input sequence of data elements is derived from MPEG encoded audio data, and wherein the decoded audio signals

10 comprise pulse code modulation samples.

8.   A method of decoding a sequence of m, m an even positive integer, input digital audio data samples $S[k]$, where $k = 0, 1, ... (m-1)$, to produce a set of n, n an even positive integer, output audio data samples $V[i]$, where $i = 0, 1, ...(n-1)$, comprising the steps of:

15 a)   calculating an array of sum data $S_{ADD}[k]$ according to

$$S_{ADD}[k] = S[k] + S[m-1-k] \qquad \text{for } k = 0, 1, ...(m/2-1)$$

b)   calculating an array of difference data $S_{SUB}[k]$ according to

$$S_{SUB}[k] = S[k] - S[m-1-k] \qquad \text{for } k = 0, 1, ...(m/2-1)$$

c)   calculating a first output audio data sample by a multiply-accumulate operation

20 according to

$$V[2i] = V[2i] + N[i, k]*S_{ADD}[k] \qquad \text{for } k = 0, 1, ... (m/2-1)$$

$$\text{where } N[i, k] = \cos\left[\frac{(32+2i)(2k+1)\pi}{64}\right]$$

d)   calculating a second output audio data sample by a multiply-accumulate operation according to

25

$$V[2i+1] = V[2i+1] + N[i, k]*S_{SUB}[k] \qquad \text{for } k = 0, 1, ... (m/2-1)$$

$$\text{where } N[i, k] = \cos\left[\frac{(32+(2i+1))(2k+1)\pi}{64}\right]$$

e)   and repeating steps c) and d) for $i = 0, 1, ... (n/2-1)$ to obtain a full set of output data.

9.   A method as claimed in claim 8, wherein the number m of input digital audio data

30 samples is 32, and the number n of output audio data samples is 32.

10. A method as claimed in claim 8 or 9, wherein the decoding steps are repeated for decoding a series of frames of encoded audio data in an MPEG format.

11. A synthesis sub-band filter for use in decoding digital audio data, comprising a means
5 for receiving or retrieving an input sequence of data elements comprising encoded digital audio data, a pre-calculation means for calculating an array of sum data and an array of difference data using selected data elements from the input sequence, and a transform calculation means for calculating a first sequence of decoded output values using said array of sum data and a second sequence of decoded output values using said array of difference
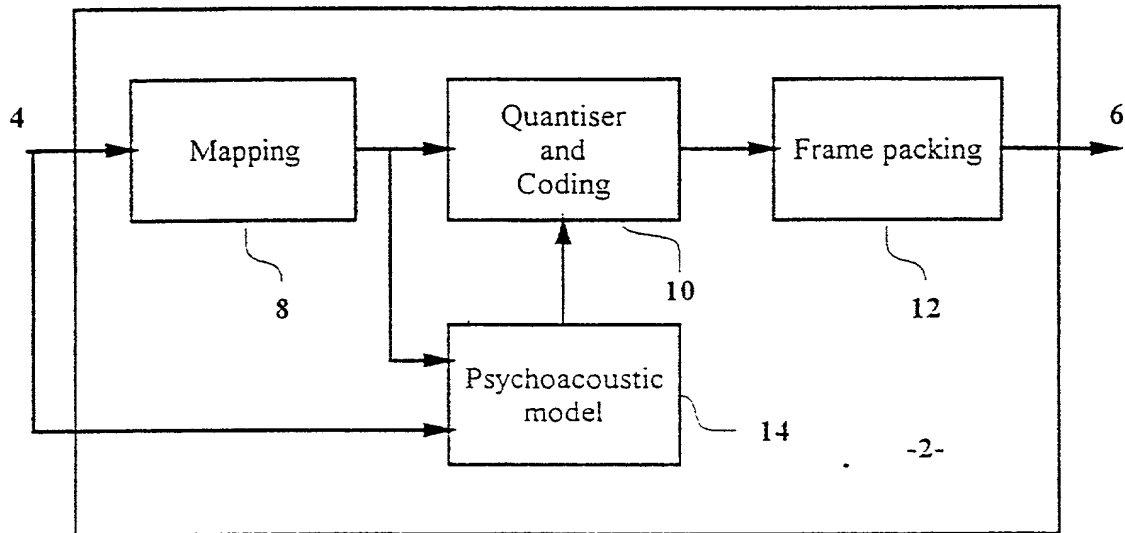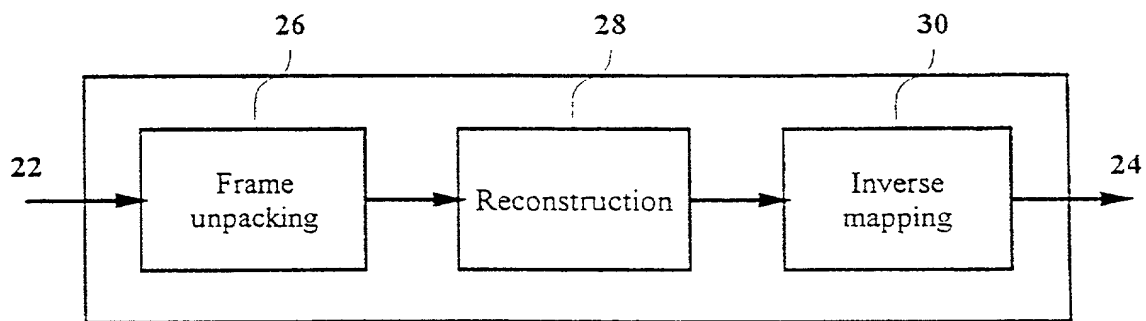10 data.

12. A synthesis sub-band filter as claimed in claim 11 wherein the pre-calculation means and transform calculation means collectively perform an inverse modified discrete cosine transform of the encoded digital audio data.
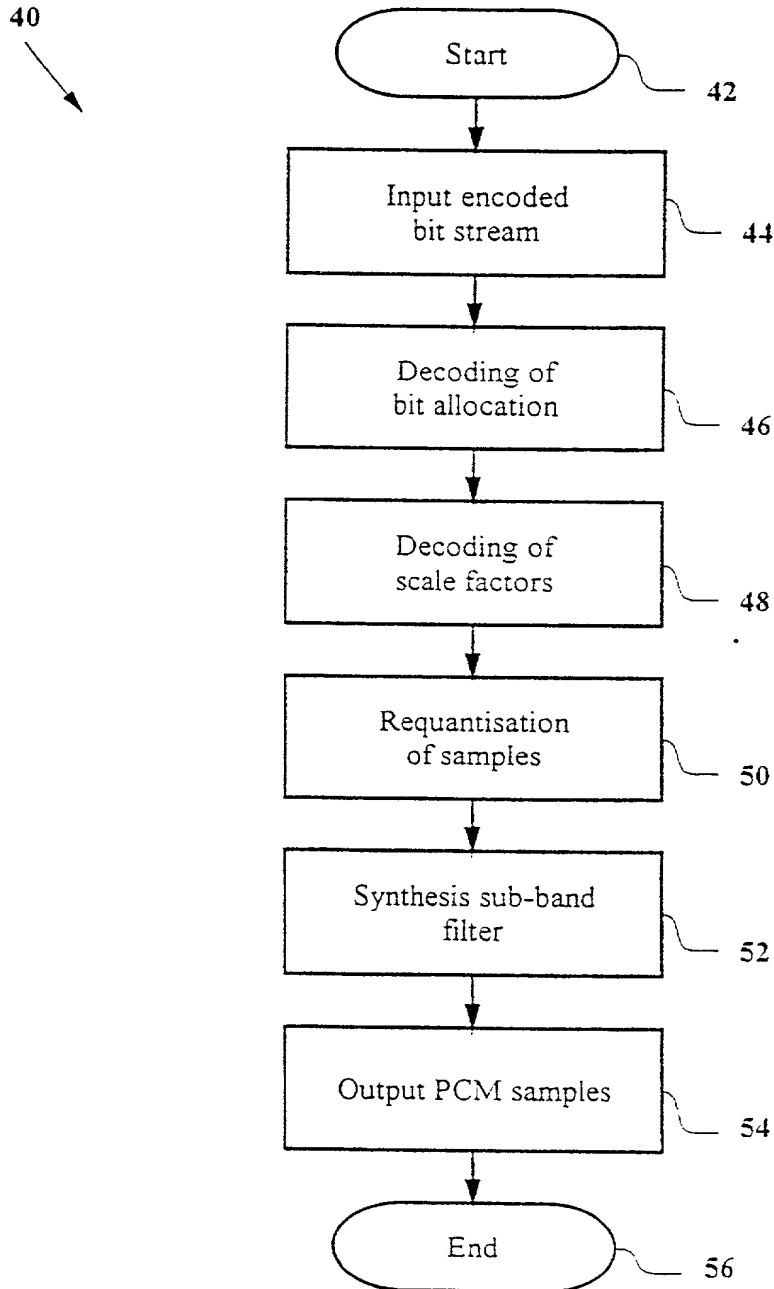
15

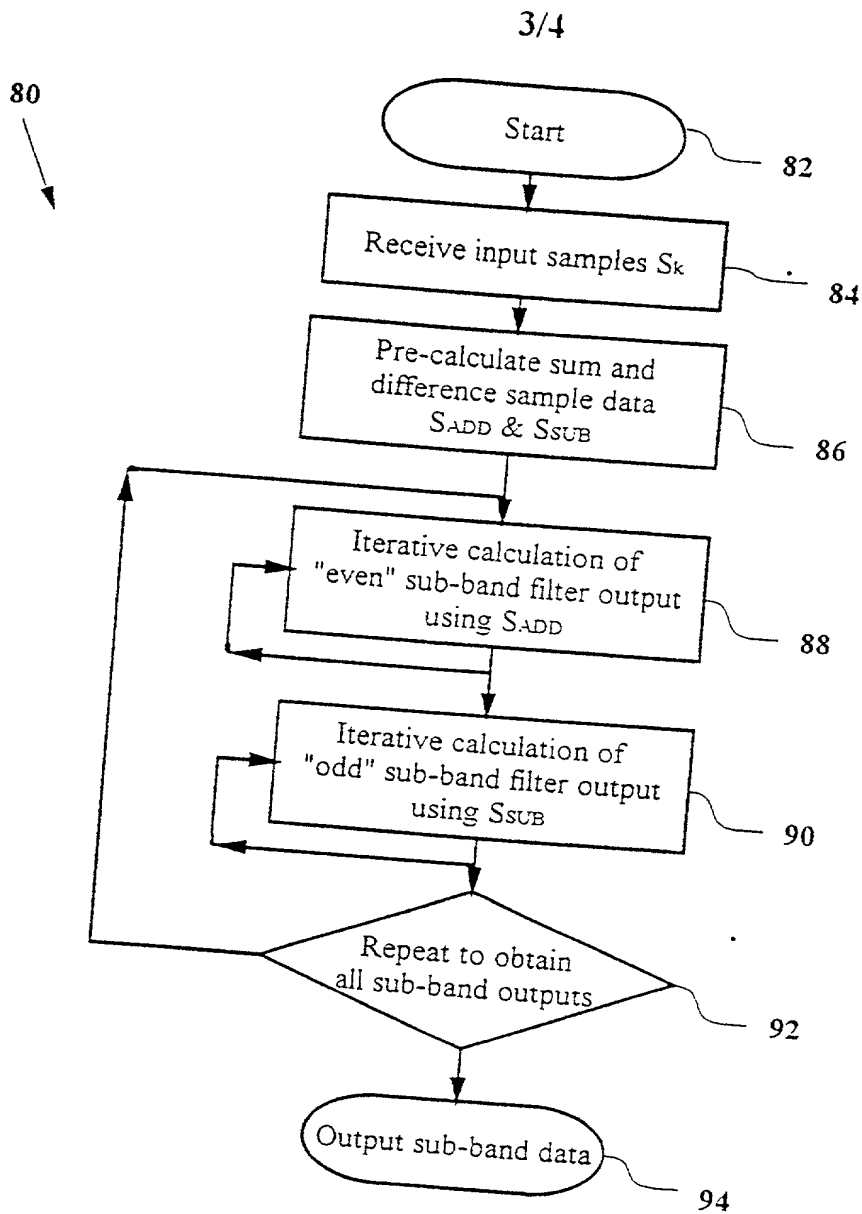13. An MPEG decoder including a synthesis sub-band filter as claimed in claim 11 or 12.

Figure 1

Figure 2

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Figure 3

80

```
              ┌──────────────┐
              │    Start     │────── 82
              └──────────────┘
                     │
                     ▼
         ┌────────────────────────┐
         │ Receive input samples Sk │────── 84
         └────────────────────────┘
                     │
                     ▼
         ┌────────────────────────┐
         │  Pre-calculate sum and  │
         │  difference sample data │
         │    S_ADD & S_SUB        │────── 86
         └────────────────────────┘
                     │
                     ▼
         ┌────────────────────────┐
         │  Iterative calculation of │
         │ "even" sub-band filter output │
         │       using S_ADD        │────── 88
         └────────────────────────┘
                     │
                     ▼
         ┌────────────────────────┐
         │  Iterative calculation of │
         │ "odd" sub-band filter output │
         │       using S_SUB        │────── 90
         └────────────────────────┘
                     │
                     ▼
            ◇────────────────◇
           ╱  Repeat to obtain  ╲
          ╱  all sub-band outputs ╲────── 92
           ╲                    ╱
            ◇────────────────◇
                     │
                     ▼
         ┌────────────────────────┐
         │  Output sub-band data    │────── 94
         └────────────────────────┘
```
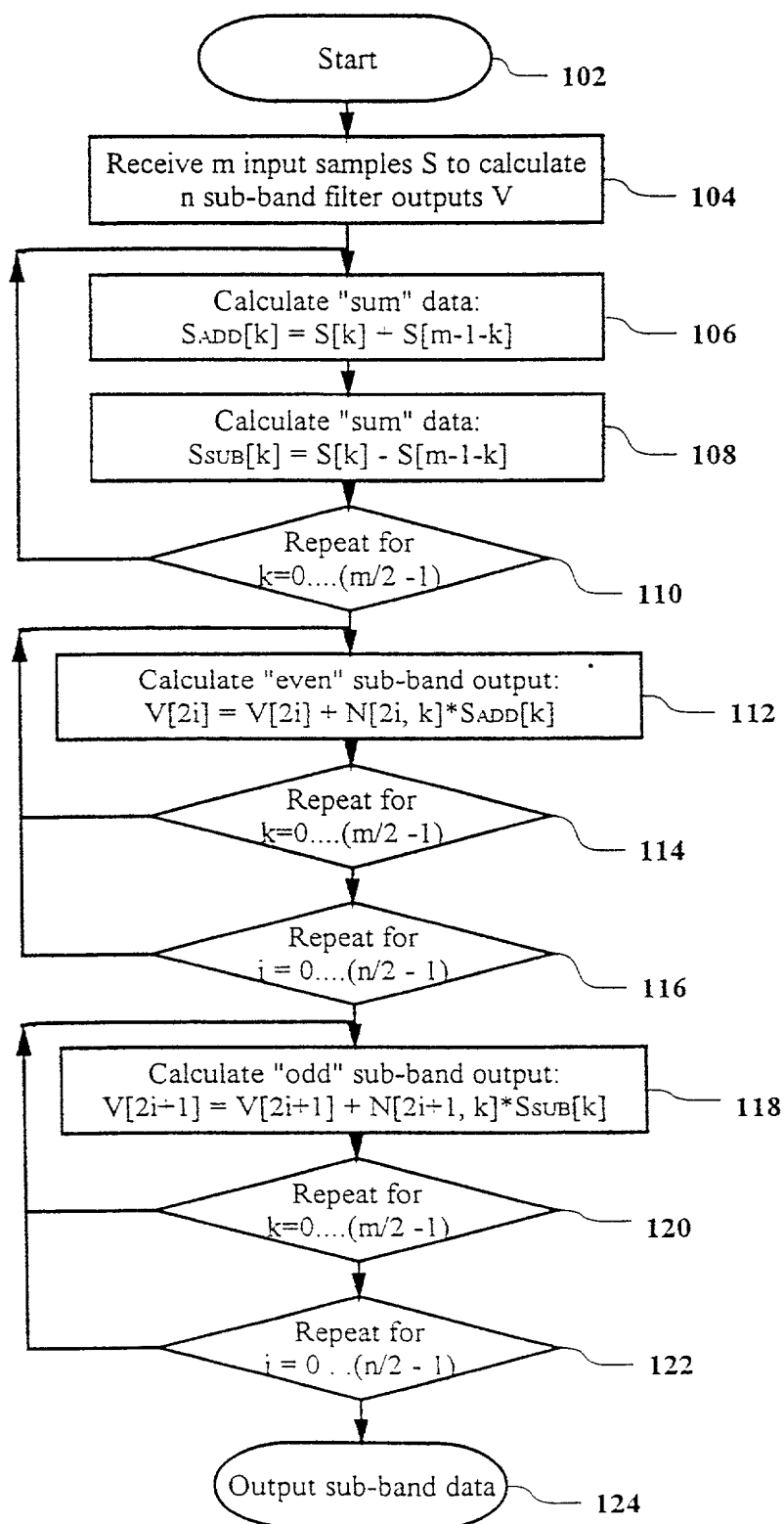
Figure 4

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Figure 5

DECLARATION AND POWER OF ATTORNEY

As the below-named inventors, we declare that:

Our residences, post office addresses, and citizenships are as stated below under our names.

We believe we are the original, first, and joint inventors of the invention entitled "FAST SYNTHESIS SUB-BAND FILTERING METHOD FOR DIGITAL SIGNAL DECODING," which is described and claimed in the specification and claims of Patent Application No. 09/486,582, which we filed in the United States Patent and Trademark Office on 28 February 2000 and for which a patent is sought.

We have reviewed and understand the contents of the foregoing specification, including the claims, as amended by any amendment specifically referred to herein (if any).

We acknowledge our duty to disclose information of which we are aware which is material to the patentability and examination of this application in accordance with 37 C.F.R. § 1.56(a).

We hereby claim foreign priority benefits under 35 U.S.C. § 119 of the foreign patent application for patent or of any PCT international application designating at least one country other than the United States of America listed below and have also identified below any foreign patent application patent or any PCT international application(s) designating at least one country other than the United States of America filed by us on the same subject matter having a filing date before that of the application on which priority is claimed:

PRIOR FOREIGN/PCT APPLICATION(S) AND ANY PRIORITY CLAIMS UNDER 35 U.S.C. 119:			
COUNTRY	APPLICATION NUMBER	DATE OF FILING	PRIORITY CLAIMED UNDER 35 USC 119
PCT	PCT/SG97/00037	29 August 1997	Yes

We hereby appoint DAVID V. CARLSON, Registration No. 31,153; MICHAEL J. DONOHUE, Reg. No. 35,859; KEVIN S. COSTANZA, Registration No. 37,801; SUSAN D. BETCHER, Reg. No. 43,498; BRIAN L. JOHNSON, Registration No. 40,033; E. RUSSELL TARLETON, Registration No. 31,800; and ROBERT IANNUCCI, Reg. No. 33,514; comprising the firm of Seed Intellectual Property Law Group PLLC, 701 Fifth Avenue, Suite 6300, Seattle, Washington 98104-7092; and LISA K. JORGENSON, Registration No. 34,845; ROBERT D. McCUTCHEON, Registration No. 38,717; JEFFREY D. MOY, Registration No. 39,307, and THEODORE E. GALANTHAY, Registration No. 24,122, as our attorneys to prosecute this application and to transact all business in the U.S. Patent and Trademark Office in connection therewith. Please direct all telephone calls to at (206) 622-4900 and telecopies to **David V. Carlson** (206) 682-6031.

We further declare that all statements made herein of our own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

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